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Amendments to the Drawings:

The attached Replacement Sheet incorporates changes made to FIG. 2. The Replacement Sheet is to replace the original drawing sheet having FIG. 2.

In FIG. 2, the term "PSTN digit map processor" (item 210) has been replaced with the term "PSTN digit string processor", and the term "VoIP digit map processor" (item 212) has been replaced with the term "VoIP digit string processor". These amendments are based on the originally filed specification, and no new matter is added.

Attachment: Replacement Sheet including FIG. 2

Annotated Sheet showing changes in FIG. 2

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REMARKS

By the above amendments, applicants have amended the specification in order to more clearly and appropriately express the subject matter thereof. Applicants have also amended claims 1, 3 and 5, and added new claims 9 and 10. Said amendments are based on the originally filed specification and drawings, and no new matter is added.

Drawings

The drawings are objected to under 37 CFR §1.83(a) because the call agent, the VoIP digit string processor, and the PSTN digit string processor must be shown or the feature(s) canceled from the claim(s).

In response

The call agent has been shown in originally filed FIG. 3 with the labels 302 and 310.

Applicants have amended FIG. 2 by replacing the term "PSTN digit map processor" with the term "PSTN digit string processor", and replacing the term "VoIP digit map processor" with the term "VoIP digit string processor". Such replacements are based on the disclosure of the original application. The PSTN digit string processor and the VoIP digit string processor, respectively labeled 210 and 212, are shown in amended FIG. 2.

Accordingly, it is submitted that the call agent, the VoIP digit string processor, and the PSTN digit string processor are shown in the drawings. Reconsideration and withdrawal of the objections are requested.

Claim Rejections – 35 USC § 112(2)

Claims 1-4 and 7 are rejected as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicants regard as the invention.

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In response

Firstly, "Call Agent" is a term commonly used in the field of VoIP (Voice over Internet Protocol). In the Media Gateway Controller Protocol (MGCP) issued as RFC2705 by the Internet Society in October 1999 (See attached), "Call Agent" is equivalent to Media Gateway Controller (MGC). According to RFC2705, the MGCP assumes a call control architecture where the call control "intelligence" is outside the telephony gateways and handled by external call control elements. "Call Agent" (See underlined in RFC2705) is such an external call control element used for controlling a telephony gateway. The telephony gateway is a network element that provides conversion between the audio signals carried on telephone circuits and data packets carried over the Internet or over other packet networks. In addition, according to RFC2705, a "Call Agent" can ask the telephony gateway to collect digits dialed by users. This facility is intended to be used with residential gateways to collect the numbers that a user dials; it may also be used with trunking gateways and access gateways alike, to collect the access codes, credit card numbers, and other numbers requested by call control services. Therefore, it is common to use a call agent to configure a VoIP digit map (see claims 3, 7). In addition, the call agent is shown in FIG. 3 with the labels 302 and 310. This illustration is consistent with the generally understood meaning of what a call agent is, as described above. Therefore it is respectfully submitted that the term call agent has a clear meaning to a person of ordinary skill in the art, and that this aspect of the claiming is definite under 35 U.S.C. §112(2).

Secondly, in claim 5, the term "PSTN digit map processor" has been replaced with the term "PSTN digit string processor", and the term "VoIP digit map processor" has been replaced with the term "VoIP digit string processor". Therefore the terminology of claim 5 is now consistent with the terminology of claim 1.

For at least the above reasons, it is submitted that claims 1-4 and 7 are now definite under 35 U.S.C. §112(2), and reconsideration and withdrawal of the rejections relating thereto are respectfully requested.

Claim Rejections – 35 USC § 103

Claims 1-8 are rejected under 35 U.S.C. §103(a) as being unpatentable over Morris (US Patent No. 7,127,043).

In response

With regard to claims 1-4:

Claim 1 recites:

"a method for a digital subscriber line device to process a dial string wherein the digital subscriber line device is coupled to a PSTN (public switched telephone network) and a VoIP (Voice-over-Internet Protocol) network, the method comprising:

receiving a transmission by the digital subscriber line device;

comparing a dial string of the transmission with phone numbers stored in a PSTN digit map and a VoIP digit map respectively;

routing the transmission to the PSTN when a phone number corresponding to the transmission is found in the PSTN digit map; and

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routing the transmission to the VoIP network when a phone number corresponding to the transmission is found in the VoIP digit map."

Morris only discloses a method of routing the transmission to the PSTN network when a phone number corresponding to the transmission is found in the PSTN digit map, and routing the transmission to the VoIP network when a phone number corresponding to the transmission is not found in the PSTN digit map. Further, **Morris fails to teach or suggest comparing the dial string of the transmission with phone numbers stored in a VoIP digit map.** In other words, Morris merely discloses a PSTN digit map and a determination as to whether a phone number corresponding to the transmission is found in the PSTN digit map or not. **The present VoIP digit map is not disclosed, taught or suggested by Morris.** Accordingly, the present step of routing the transmission to the VoIP network when a phone number corresponding to the transmission is found in the VoIP digit map is not disclosed, taught or suggested.

In addition, claim 1 can more correctly determine whether a dial string should be routed to the PSTN network or to the VoIP network by comparing the dial string of the transmission with both types of digit maps. The method of claim 1 is more reliable and convenient than that of Morris, which does not teach or suggest these new and unexpected results.

For at least the above reasons, it is submitted that Morris does not disclose, teach or otherwise suggest the invention as currently set forth in claim 1. Claim 1 is novel, unobvious and patentable over Morris under both 35 U.S.C. §102 and 35 U.S.C. §103(a).

Claims 2 to 4 are directly dependent on claim 1, and incorporate more features

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therein respectively. Therefore claims 2 to 4 should also be patentable.

With regard to claims 5-8:

Claim 5 recites a digital subscriber line device including a PSTN digit string processor and a VoIP digit string processor for comparing a dial string of the transmission with phone numbers respectively stored in a PSTN digit map and a VoIP digit map. The PSTN digit string processor routes the transmission to the PSTN when a phone number corresponding to the transmission is found in the PSTN digit map. The VoIP digit string processor routes the transmission to the VoIP network when a phone number corresponding to the transmission is found in the VoIP digit map.

Morris only discloses a system of routing the transmission to the PSTN network when a phone number corresponding to the transmission is found in the PSTN digit map, and routing the transmission to the VoIP network when a phone number corresponding to the transmission is not found in the PSTN digit map. However, **Morris fails to teach or suggest comparing the dial string of the transmission with phone numbers stored in a VoIP digit map.** In other words, Morris merely discloses a PSTN digit map and determination as to whether a phone number corresponding to the transmission is found in the PSTN digit map or not. **The present VoIP digit map is not disclosed, taught or suggested by Morris.** Accordingly, the present VoIP digit string processor that routes the transmission to the VoIP network when a phone number corresponding to the transmission is found in the VoIP digit map is not disclosed, taught or suggested.

In addition, claim 5 can more correctly determine whether a dial string should be

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routed to the PSTN network or the VoIP network by comparing the dial string of the transmission with both types of digit maps. The digital subscriber line device of claim 5 is more reliable and convenient than the system of Morris, which does not teach or suggest these new and unexpected results.

For at least the above reasons, it is submitted that Morris does not disclose, teach or otherwise suggest the invention as currently set forth in claim 5. Claim 5 is novel, unobvious and patentable over Morris under both 35 U.S.C. §102 and 35 U.S.C. §103(a).

Claims 6 to 8 are directly dependent on claim 5, and incorporate more features therein respectively. Therefore claims 6 to 8 should also be patentable.

New claims – 35 USC § 103

Claim 9 is directly dependent on claim 1, which is submitted to be patentable for the reasons stated above. Therefore claim 9 should also be patentable.

If further argument is needed, claim 9 recites "transmitting a voice signal to the telephone to notify the user of a dial error message when a phone number corresponding to the transmission is neither found in the PSTN digit map nor in the VoIP digit map".

As detailed above, Morris does not disclose, teach or suggest anything in relation to the present VoIP digit map. Further, the method of claim 9 yields new and unexpected results. In particular, the use of the two types of digit maps provides the basis for the claimed error notification process. Referring also to the eleventh paragraph of the Detailed Description Of The Invention (on p.8 of the specification), the method provides transmission of voice signals to the telephone to notify the user of a dial error message when a phone number corresponding to the transmission is neither found in the PSTN

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digit map nor in the VoIP digit map. In contrast, Morris cannot give notice of a dialing error due to there being only one type of digit map available. When a person dials a wrong number, Morris only fails to find a match for the dial string in the PSTN digit map, and accordingly routes the call to the VoIP network. Thus Morris's method is inconvenient for a user, with no teaching or suggestion of the reliability and convenience provided by the present method.

For at least the above reasons, it is submitted that claim 9 is patentable.

Claim 10 is directly dependent on claim 5, which is submitted to be patentable for the reasons stated above. Therefore claim 10 should also be patentable.

If further argument is needed, claim 10 recites "an error notification module for transmitting a voice signal to the telephone to notify the user of a dial error message when a phone number corresponding to the transmission is neither found in the PSTN digit map nor in the VoIP digit map".

As detailed above, Morris does not disclose, teach or suggest anything in relation to the present VoIP digit map. Further, the digital subscriber line device of claim 10 yields new and unexpected results. In particular, the use of the two types of digit maps provides the basis for the claimed error notification mechanism. Referring also to the eleventh paragraph of the Detailed Description Of The Invention (on p.8 of the specification), the error notification module provides transmission of voice signals to the telephone to notify the user of a dial error message when a phone number corresponding to the transmission is neither found in the PSTN digit map nor in the VoIP digit map. In contrast, Morris cannot give notice of a dialing error due to there being only one type of

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digit map available. When a person dials a wrong number, Morris only fails to find a match for the dial string in the PSTN digit map, and accordingly routes the call to the VoIP network. Thus Morris's method is inconvenient for a user, with no teaching or suggestion of the reliability and convenience provided by the present digital subscriber line device.

For at least the above reasons, it is submitted that claim 10 is patentable.

CONCLUSION

Applicants submit that the foregoing Amendment and Response place this application in condition for allowance. If Examiner believes that there are any issues that can be resolved by a telephone conference, or that there are any informalities that can be corrected by an Examiner's amendment, please call the undersigned at 714.626.1240.

Respectfully,
Ming-Chieh Chen,
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By ____________

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Network Working Group
Request for Comments: 2705
Category: Informational

M. Arango
RSL COM
A. Dugan
I. Elliott
Level3 Communications
C. Huitema
Telcordia
S. Pickett
Vertical Networks
October 1999

Media Gateway Control Protocol (MGCP)

Version 1.0

Status of this Memo

This memo provides information for the Internet community. It does not specify an Internet standard of any kind. Distribution of this memo is unlimited.

Copyright Notice

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IESG NOTE:

This document is being published for the information of the community. It describes a protocol that is currently being deployed in a number of products. Implementers should be aware of developments in the IETF Megaco Working Group and ITF-T SG16 who are

currently working on a potential successor to this protocol.

Abstract

This document describes an application programming interface and a corresponding protocol (MGCP) for controlling Voice over IP (VoIP) Gateways from external call control elements. MGCP assumes a call control architecture where the call control "intelligence" is outside the gateways and handled by external call control elements.

The document is structured in 6 main sections:

* The introduction presents the basic assumptions and the relation to other protocols such as H.323, RTSP, SAP or SIP.
* The interface section presents a conceptual overview of the MGCP, presenting the naming conventions, the usage of the session description protocol SDP, and the procedures that compose MGCP: Notifications Request, Notification, Create Connection, Modify Connection, Delete Connection, AuditEndpoint, AuditConnection and RestartInProgress.
* The protocol description section presents the MGCP encodings, which are based on simple text formats, and the transmission procedure over UDP.
* The security section presents the security requirement of MGCP, and its usage of IP security services (IPSEC).
* The event packages section provides an initial definition of packages and event names.

* The description of the changes made in combining SGCP 1.1 and IPDC to create MGCP 1.0.

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1. Introduction

This document describes an abstract application programming interface and a corresponding protocol (MGCP) for controlling Telephony Gateways from external call control elements called media gateway controllers or call agents. A telephony gateway is a network element that provides conversion between the audio signals carried on telephone circuits and data packets carried over the Internet or over other packet networks. Example of gateways are:

* Trunking gateways, that interface between the telephone network and a Voice over IP network. Such gateways typically manage a large number of digital circuits.

* Voice over ATM gateways, which operate much the same way as voice over IP trunking gateways, except that they interface to an ATM network.

* Residential gateways, that provide a traditional analog (RJ11) interface to a Voice over IP network. Examples of residential gateways include cable modem/cable set-top boxes, xDSL devices, broad-band wireless devices

* Access gateways, that provide a traditional analog (RJ11) or digital PBX interface to a Voice over IP network. Examples of access gateways include small-scale voice over IP gateways.

* Business gateways, that provide a traditional digital PBX interface or an integrated "soft PBX" interface to a Voice over IP network.

* Network Access Servers, that can attach a "modem" to a telephone circuit and provide data access to the Internet. We expect that, in the future, the same gateways will combine Voice over IP services and Network Access services.

* Circuit switches, or packet switches, which can offer a control interface to an external call control element.

MGCP assumes a call control architecture where the call control "intelligence" is outside the gateways and handled by external call control elements. The MGCP assumes that these call control elements, or <u>Call Agents,</u> will synchronize with each other to send coherent

commands to the gateways under their control. MGCP does not define a mechanism for synchronizing Call Agents. MGCP is, in essence, a master/slave protocol, where the gateways are expected to execute commands sent by the Call Agents. In consequence, this document specifies in great detail the expected behavior of the gateways, but only specify those parts of a call agent implementation, such as timer management, that are mandated for proper operation of the protocol.

MGCP assumes a connection model where the basic constructs are endpoints and connections. Endpoints are sources or sinks of data and could be physical or virtual. Examples of physical endpoints are:

* An interface on a gateway that terminates a trunk connected to a PSTN switch (e.g., Class 5, Class 4, etc.). A gateway that terminates trunks is called a trunk gateway.

* An interface on a gateway that terminates an analog POTS connection to a phone, key system, PBX, etc. A gateway that terminates residential POTS lines (to phones) is called a residential gateway.

An example of a virtual endpoint is an audio source in an audio-content server. Creation of physical endpoints requires hardware installation, while creation of virtual endpoints can be done by software.

Connections may be either point to point or multipoint. A point to point connection is an association between two endpoints with the purpose of transmitting data between these endpoints. Once this

association is established for both endpoints, data transfer between these endpoints can take place. A multipoint connection is established by connecting the endpoint to a multipoint session.

Connections can be established over several types of bearer networks:

* Transmission of audio packets using RTP and UDP over a TCP/IP network.
* Transmission of audio packets using AAL2, or another adaptation layer, over an ATM network.
* Transmission of packets over an internal connection, for example the TDM backplane or the interconnection bus of a gateway. This is used, in particular, for "hairpin" connections, connections that terminate in a gateway but are immediately rerouted over the telephone network.

For point-to-point connections the endpoints of a connection could be in separate gateways or in the same gateway.

2.1.5. Digit maps

The Call Agent can ask the gateway to collect digits dialed by the user. This facility is intended to be used with residential gateways to collect the numbers that a user dials; it may also be used with trunking gateways and access gateways alike, to collect the access codes, credit card numbers and other numbers requested by call control services.

An alternative procedure is for the gateway to notify the Call Agent of the dialed digits, as soon as they are dialed. However, such a procedure generates a large number of interactions. It is preferable to accumulate the dialed numbers in a buffer, and to transmit them in a single message.

The problem with this accumulation approach, however, is that it is hard for the gateway to predict how many numbers it needs to accumulate before transmission. For example, using the phone on our desk, we can dial the following numbers:

0	Local operator
00	Long distance operator
xxxx	Local extension number
8xxxxxxx	Local number
#xxxxxxx	Shortcut to local number at other corporate sites
*xx	Star services
91xxxxxxxxxx	Long distance number
9011 + up to 15 digits	International number

The solution to this problem is to load the gateway with a digit map that correspond to the dial plan. This digit map is expressed using a syntax derived from the Unix system command, egrep. For example, the dial plan described above results in the following digit map:

(0T| 00T|[1-7]xxx|8xxxxxxx|#xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)

The formal syntax of the digit map is described by the DigitMap rule in the formal syntax description of the protocol (section 3.4). A Digit-Map, according to this syntax, is defined either by a "string" or by a list of strings. Each string in the list is an alternative numbering scheme, specified either as a set of digits or timers, or as regular expression. A gateway that detects digits, letters or timers will:

1) Add the event parameter code as a token to the end of an internal state variable called the "current dial string"

2) Apply the current dial string to the digit map table, attempting a match to each regular expression in the Digit Map in lexical order

3) If the result is under-qualified (partially matches at least one entry in the digit map), do nothing further.

If the result matches, or is over-qualified (i.e. no further digits could possibly produce a match), send the current digit string to the Call Agent. A match, in this specification, can be either a "perfect match," exactly matching one of the specified alternatives, or an impossible match, which occur when the dial string does not match any of the alternative. Unexpected timers, for example, can cause "impossible matches." Both perfect matches and impossible matches trigger notification of the accumulated digits.

Digit maps are provided to the gateway by the Call Agent, whenever the Call Agent instructs the gateway to listen for digits.

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Amdt. Dated October 18, 2007
Reply to Office action of July 18, 2007

Annotated Sheet

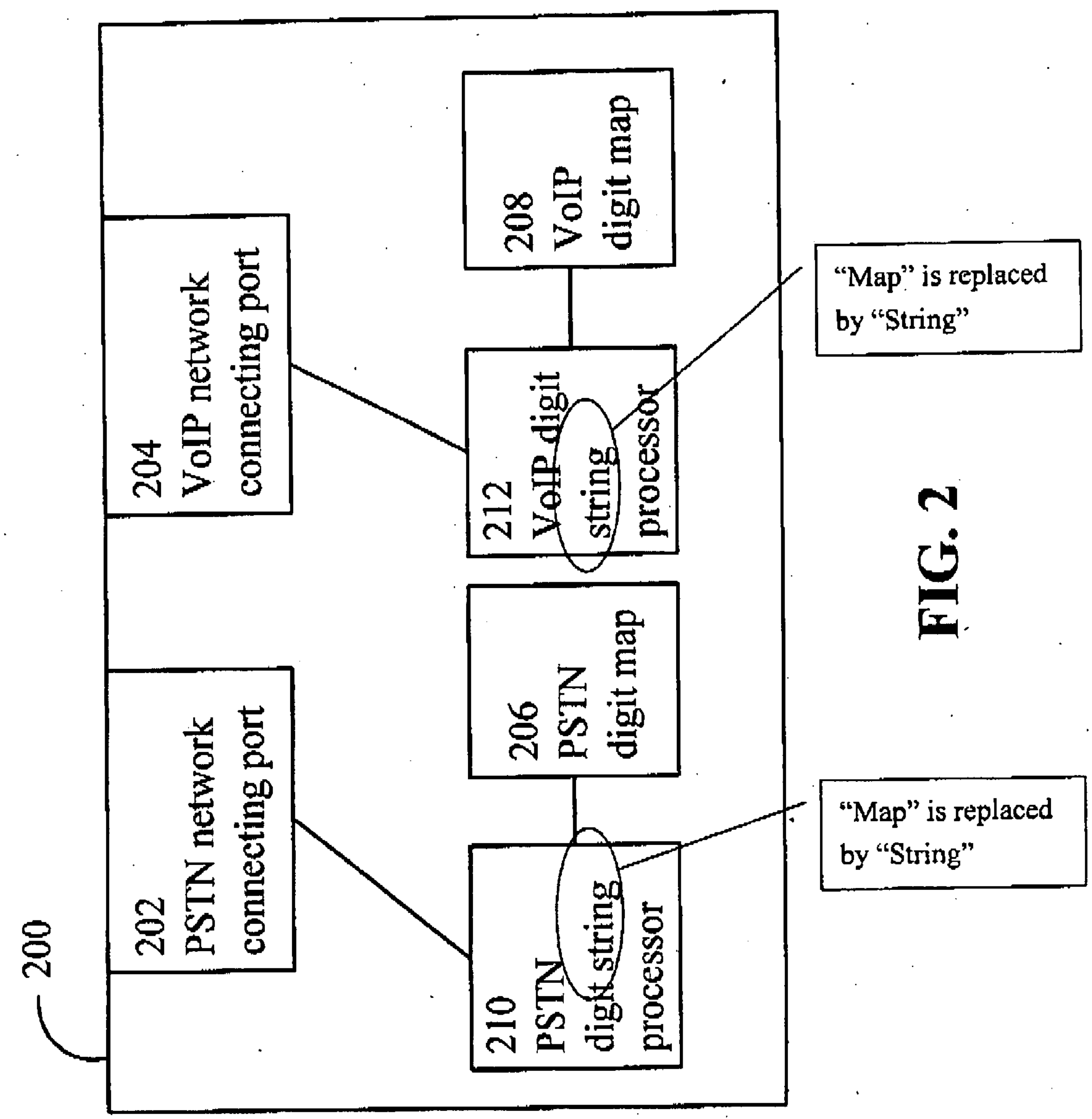

FIG. 2

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